Amendments to the Claims

The listing of claims will replace all prior versions, and listings of claims in the application.

1. (*currently amended*) A method for reducing overhead, latency, and/or packet loss for ~~and latency and handling packet loss in~~ a voice and data over Internet Protocol (VoIP) ~~data packet, transmitted between originating and destination gateways in an Internet telephony system~~ packet transmitted between an originating gateway and a destination gateway, comprising the steps of:

(1) compressing voice streams and/or data streams from a plurality of concurrent calls from a plurality of channels into packets;

(2) aggregating said packets into ~~the larger data~~ a single packet to produce the VoIP packet, ~~said data packet~~ said VoIP packet including information ~~for synchronizing~~ , that when executed, synchronizes a current channel state at the originating gateway with a record of said channel state at the destination gateway; and

(3) transmitting the ~~data packet~~ VoIP packet between the originating gateway and the destination ~~gateways~~ gateway through a single virtual connection.

2. (*currently amended*) The method of claim 1, wherein step (2) further comprises the step of providing a plurality of voice frames and/or data frames and a plurality of header frames in the ~~data packet~~ VoIP packet, wherein said plurality of header frames comprises at least one header frame selected from the group consisting of a time stamp header, local network address header, IP address header and UDP header and at least one header frame selected from the group consisting of a version number header and control information header.

3. (*currently amended*) The method of claim 1, ~~wherein step (1) further comprises~~ further comprising the step of converting analog voice streams and/or analog data streams to digital voice streams and/or digital data streams prior to executing said compressing ~~said data streams into said packets~~ step.

4. (*currently amended*) The method of claim 1, further comprising the step of transmitting a check sequence [data] packet at regular packet intervals, wherein [the] a duration of said intervals is altered to reach a desired tradeoff between increased tolerance to loss and bandwidth, wherein a parity system and [the] information located inside of said check sequence [data] packet is used to regenerate missing or damaged [data] information in [the] a previously transmitted ~~data packet~~ VoIP packet.

5. (*currently amended*) A method for regenerating missing or damaged ~~data in a data packet transmitted in an Internet telephony system~~ information in a VoIP packet, comprising the steps of:

~~(1) transmitting a check sequence data packet after the transmission of every third data packet, wherein information located inside of said check sequence data packet is used to regenerate missing or damaged data in any of the preceding three data packets; and~~

~~(2) using a parity system to regenerate the missing or damaged data~~

(1) transmitting a plurality of VoIP packets from a source to a destination; and

(2) transmitting a check sequence packet from said source to said destination upon completion of a transmission of said plurality of VoIP packets, wherein said check sequence packet comprises information that, when executed, regenerates missing or damaged information transmitted in any of said plurality of VoIP packets.

6. (*currently amended*) A system for reducing overhead, latency, and/or ~~and latency and handling~~ packet loss in a voice and data over Internet Protocol (VoIP) ~~data packet, transmitted over a UDP/IP connectionless protocol between originating and destination gateways~~ packet transmitted between an originating gateway and a destination gateway, said system comprising:

media framing means for aggregating packets from a plurality of concurrent calls from a plurality of channels into ~~the larger data~~ a single packet to produce a VoIP packet;

transmission control means for providing information in the ~~data packet~~ VoIP packet to synchronize a current channel state at the originating gateway with a record of said channel state at the destination gateway;

redundancy means for regenerating missing or damaged ~~data in the data packet~~ information transmitted in the VoIP packet; and

a single virtual connecting means for transmitting the ~~data packet~~ VoIP packet from the originating gateway to the destination gateway.

7. (*currently amended*) The system of claim 6, wherein the ~~data packet~~ VoIP packet comprises a plurality of voice frames and/or data frames and a plurality of header frames, comprising at least one header frame selected from the group consisting

of a time stamp header, local network address header, IP address header and UDP header and at least one header frame selected from the group consisting of a version number header and control information header.

8. (*currently amended*) The system of claim 6, further comprising:

means for transmitting and receiving voice streams and/or data streams;

means for converting analog voice streams and/or analog data streams to digital voice streams and/or digital data streams;

means for compressing digital voice streams and/or digital data streams into said packets; and

means for transmitting a check sequence ~~data packet after the transmission of every third data packet~~ packet after a transmission of a predetermined quantity of VoIP packets, wherein said media framing means produces each VoIP packet of said predetermined quantity of VoIP packets.

9. (*currently amended*) The system of claim 8, wherein said ~~check sequence data packet is formatted to regenerate said missing or damaged data with information located inside of said check sequence data packet, and use a parity system to regenerate said missing or damaged data~~ check sequence packet includes check sequence information that, when executed, regenerates missing or damaged information transmitted in the VoIP packet, wherein said redundancy means produces said check sequence information.

10. (*currently amended*) An Internet telephony system for regenerating missing or damaged ~~data in a data packet~~ information in a VoIP packet, comprising:

redundancy means for transmitting a check sequence ~~data packet after every three or more data packets~~ packet upon completion of a transmission of a predetermined quantity of VoIP packets; and

means for regenerating ~~the missing or damaged data with the information located inside of said check sequence data packet~~ missing or damaged information in any of said predetermined quantity of VoIP packets.

11. (*currently amended*) The system of claim 10, further comprising means for implementing a parity system to regenerate said ~~missing or damaged data~~ missing or damaged information.

12. (*currently amended*) A computer program product comprising a computer useable medium having computer ~~program logic recorded thereon for enabling originating and destination gateways to transmit or receive data streams or data packets in an Internet telephony system and for reducing VoIP packet overhead and latency and handling packet loss~~ readable program code means embedded in said medium for causing a computer to reduce overhead, latency, and/or packet loss in a VoIP packet transmitted between an originating gateway and a destination gateway, ~~said computer program logic~~ comprising:

~~a first computer program product means for compressing the~~ first computer readable program code means for causing the computer to compress voice

streams and/or data streams from a plurality of concurrent calls from a plurality of channels into packets;

~~a second computer program product means for aggregating~~ second computer readable program code means for causing the computer to aggregate said packets into ~~the larger data packets~~ a single packet to produce the VoIP packet;

~~a third computer program product means for transmitting the data packets~~ third computer readable program code means for causing the computer to transmit the VoIP packet between the originating gateway and the destination ~~gateways~~ gateway through a single virtual connection;

~~a fourth computer program product means for providing~~ fourth computer readable program code means for causing the computer to provide information in the ~~data packets~~ VoIP packet to synchronize a current channel state at the originating gateway with a record of said channel state at the destination gateway; and

~~a fifth computer program product means for determining~~ fifth computer readable program code means for causing the computer to determine if the ~~data packets contain missing or damaged data and regenerating said missing or damaged data in the data packets~~ VoIP packet contains missing or damaged information or to regenerate said missing or damaged information.

13. (*currently amended*) The computer program product of claim 12, wherein said second computer ~~program product means~~ readable program code means further comprises computer ~~program product means for aggregating~~ readable program code means for causing the computer to provide ~~packets into the data packets comprising~~ a plurality of voice frames and/or data frames and a plurality of header frames in the VoIP

packet, wherein said plurality of header frames comprises at least one header frame selected from the group consisting of a time stamp header, local network address header, IP address header and UDP header and at least one header frame selected from the group consisting of a version number header and control information header.

14. (*currently amended*) The computer program product of claim 12, ~~wherein said first computer program product means further comprises computer program product means for converting~~ further comprising sixth computer readable program code means for causing the computer to convert analog voice streams and/or analog data streams to digital voice streams and/or digital data streams prior to ~~compressing the data streams into said packets~~ executing said first computer readable program code means.

15. (*currently amended*) The computer program product of ~~claim 14~~ claim 12, wherein said fifth computer ~~program product means~~ readable program code means further comprises computer ~~program product means for transmitting a check sequence data packet after every three data packets and using a parity system and the information located inside of said check sequence data packet to regenerate said missing or damaged data~~ readable program code means for causing the computer to transmit a check sequence packet upon completion of an execution of said third computer readable program code means, wherein said check sequence packet comprises information that, when executed, regenerates said missing or damaged information.

16. (*currently amended*) A computer program product comprising a computer useable medium having computer ~~program logic recorded thereon for enabling~~

~~originating and destination gateways to transmit or receive data streams or data packets in an Internet telephony system and for regenerating missing or damaged data in the data packets~~ <u>readable program code means embedded in said medium for causing a computer to regenerate missing or damaged information in a VoIP packet transmitted between an originating gateway and a destination gateway</u>, comprising:

~~a first computer program product means for transmitting~~ <u>first computer readable program code means for causing the computer to transmit</u> a check sequence [data] packet at regular packet intervals, wherein ~~the duration of said intervals is altered~~ <u>said first computer readable program code means comprises computer readable program code means for causing the computer to alter a duration of said intervals</u> to reach a desired tradeoff between increased tolerance to loss and bandwidth; and

~~a second computer program product means for regenerating the~~ <u>second computer readable program code means for causing the computer to regenerate</u> missing or damaged [data] <u>information</u> in a previously transmitted [data] <u>VoIP</u> packet by using information located inside of said check sequence [data] packet.

17. (*currently amended*) The computer program product of claim 16, further comprising ~~a third computer program product means for using~~ <u>third computer readable program code means for causing the computer to utilize</u> a parity system to regenerate [the] <u>said</u> missing or damaged [data] <u>information</u>.

18. (*previously presented*) The method of claim 1, wherein said channel state identifies whether a channel is open or on-line.

19. (*currently amended*) The method of claim 1, wherein step (2) further comprises the step of providing in the [data] VoIP packet a channel present header for indicating whether a channel is currently open and communicating.

20. (*currently amended*) The method of claim 1, wherein step (2) further comprises the step of providing information in the [data] VoIP packet to instruct the destination gateway to start using said record to deframe the [data] VoIP packet.

21. (*currently amended*) The system of claim 6, wherein said single virtual connecting means enables transmission of the [data] VoIP packet from said media framing means at the originating gateway directly to a second media framing means at the destination gateway.

22. (*currently amended*) The system of claim 6, wherein said single virtual connecting means enables transmission of the [data] VoIP packet from said transmission control means at the originating gateway directly to a second transmission control means at the destination gateway.